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Title of Invention	:	METHOD AND APPARATUS FOR GENERATING DIRECTIONAL SOUND USING AN ULTRASONIC CARRIER WAVE

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THE PATENTS ACT
(CHAPTER 221)

CERTIFICATE OF GRANT OF PATENT

In accordance with section 35 of the Patents Act, it is hereby certified that a patent having the P-No. 114498 has been granted in respect of an invention having the following particulars:

Title	:	METHOD AND APPARATUS FOR GENERATING DIRECTIONAL SOUND USING AN ULTRASONIC CARRIER WAVE
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Name of Inventor(s)	:	WOON-SENG GAN ; MENG-HWA ER ; YEW HIN LIEW ; CHEE-MUN LEE ; FURI ANDI KARNAPI
Name(s) and Address(es) of Proprietor(s) of Patent	:	NANYANG TECHNOLOGICAL UNIVERSITY BLOCK S2, SCHOOL OF ELECTRICAL AND ELECTRONIC ENGINEERING NANYANG AVE SINGAPORE 639798
Date of Grant	:	30 December 2005

Dated this 30th day of December 2005.

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ACTION

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METHOD AND APPARATUS FOR GENERATING DIRECTIONAL SOUND USING AN ULTRASONIC CARRIER WAVE

INVENTORS:

Woon-Seng Gan, Meng-Hwa Er, Yew-Hin Liew, Chee-Mun Lee, Furi Andi Karnapi

BACKGROUND OF THE INVENTION

Audio signals are normally provided to listeners via an audio path that includes a source (e.g. a CD, tape cassette player, microphone etc.), an amplifier, and an electrodynamic cone-type loudspeaker. More recently, interest has been shown in delivering audio using an ultrasonic carrier wave. In this case, an audio-frequency signal is amplitude-modulated onto the ultrasonic carrier wave. The modulated carrier wave is then provided to an ultrasonic loudspeaker for delivery. Demodulation of the audio-frequency signal either occurs naturally as a result of the interaction of the carrier wave with the transmission medium (normally air, which has non-linear characteristics of finite amplitude propagation for ultrasonic waves). Reflection can, for example, take place when the modulated carrier wave encounters an object that absorbs energy at ultrasonic frequencies but reflects energy at audio frequencies.

The advantages of using an ultrasonic carrier wave to deliver audio include the highly directional nature of an ultrasonic wave, the fact that the carrier wave is steerable (for example by providing reflective surfaces), and also that the signal is not audible prior to demodulation. By proper application of these advantages, audio can be delivered to specific locations, from where the audio appears to originate. A general discussion of the

transmission of audio signals can be found in European published patent application no. EP 973 152.

However, technical challenges remain in the use of ultrasonic technology for delivering audio. For example, the fidelity of the demodulated audio signal can still be improved.

In "Audio Application of the Parametric Array," J. Acoust. Soc. Am., Vol 102 pp 3106(A), 1997, J. Blackstock suggests a method of improving distortion in the demodulated signal. In particular, it is noted that, when the primary wave is a modulated carrier, the sound generated (upon demodulation) by the secondary (modulating) wave is proportional to the second time derivative of the square of the modulation envelope. This results in high levels of harmonic distortion in the sound generated. To overcome this, Blackstock proposes integrating the original signal twice and taking the square root thereof, to anticipate the demodulation function and thus remove the distortion resulting from demodulation. This is shown in the following equation, where f(t) is the audio signal and E(t) is the signal provided to the modulator:

$$E(t) = \left(1 + \iint f(t)dt^2\right)^{1/2}$$

This preprocessing approach however, due to the square root operation, generates an infinite number of harmonics. In other words, the harmonic distortion will only be removed if all of these harmonics are reproduced. Thus, the amount of distortion of the demodulated signal is directly related to the bandwidth of the device. That is, this method requires bandwidth-intensive ultrasonic path and emitters to get optimal performance.

SUMMARY OF INVENTION

According to one aspect of the invention there is provided a method of processing an audio signal, comprising:

performing a square root operation on the audio signal to generate a square rooted signal;

alternating the gain of the square rooted signal between positive and negative gain values at selected locations to generate a flipped signal; and

modulating the flipped signal onto a first ultrasonic carrier wave.

Preferably the audio signal is offset by a predetermined amount prior to performing the square root operation to ensure that the square root operation only results in real values.

The method also preferably includes:

dividing the audio signal into a plurality of frames;

determining, after the offsetting step; a minimum value of a portion of the audio signal in a particular frame; and

subtracting the minimum value from the portion of the audio signal in the particular frame.

In the preferred embodiment, the selected locations between which the signal is flipped are minimum turning points of the signal.

Still further, the method may comprise the steps of:

determining a first modulation envelope for the flipped signal;

determining a second modulation envelope for the square rooted signal;

determining the difference between the first and second modulation envelopes;

modulating the difference between the first and second modulation envelopes onto a second ultrasonic carrier wave.

Preferably, the first and second ultrasonic carrier waves are orthogonal to one another.

According to another aspect of the invention, there is provided an apparatus for processing an audio signal received from an audio source. The apparatus preferably comprises:

a square root module to perform a square root operation on the audio signal to generate a square rooted signal;

a determining module coupled to the square root module to alternate the gain of the square rooted signal between positive and negative gain values at selected locations thereby to generate a flipped signal; and

a modulator to modulate the flipped signal onto a first ultrasonic carrier wave.

The apparatus may further comprise:

an offset module to offset the audio signal by a predetermined amount prior to passing the signal to the square root module.

Yet further, the apparatus may comprise:

a buffer to divide the audio signal into a plurality of frames;

a subtracting module to subtract a minimum value from the portion of the audio signal in the particular frame.

Preferably, the determining module also

determines a first modulation envelope for the flipped signal;

determines a second modulation envelope for the square rooted signal;

determines the difference between the first and second modulation envelopes;

and the modulator

modulates the difference between the first and second modulation envelopes onto a second ultrasonic carrier wave.

According to another aspect of the invention, there is provided a method for processing an audio signal received from an audio source, comprising:

processing the audio signal into a first processed audio signal;

processing the audio signal into a second processed audio signal;

modulating the first processed audio signal onto a first ultrasonic carrier wave; and

modulating the second processed audio signal onto a second ultrasonic carrier wave;

wherein the first and second ultrasonic carrier waves have different phases.

Preferably, the first ultrasonic carrier wave is orthogonal to the second ultrasonic carrier wave.

Still further, according to another aspect of the invention there is provided an apparatus for processing an audio signal received from an audio source, comprising:

a processor to process the audio signal into a first processed audio signal and a second processed audio signal;

a modulator to modulate the first processed audio signal onto a first ultrasonic carrier wave and to modulate the second processed audio signal onto a second ultrasonic carrier wave;

wherein the first and second ultrasonic carrier waves have different phases.

Preferably, the first ultrasonic carrier wave is orthogonal to the second ultrasonic carrier wave.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will now be described in more detail with reference to the attached figures, in which:

FIG. 1 is a schematic diagram of an ultrasonic signal processing system according to one aspect of the invention;

FIG. 2 is a schematic diagram of the preprocessor of the system of Fig. 1;

FIG. 3 shows two curves illustrating part of the processing done in the system of Figs 1 and 2;

FIG. 4 is a schematic diagram of an ultrasonic emitter suitable for use with the system of Fig. 1.

FIG. 5 is a schematic diagram of the modulator of the system of Fig. 1.; and

FIG. 6 shows two frequency response graphs comparing the modified square root method of the invention with the traditional square root method;

FIG. 7 shows two modulated waveform graphs comparing the modified square root of the invention with the traditional square root method for a 1 kHz, 2 kHz and 4 kHz modulating waveform.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

FIG. 1 is a schematic diagram that shows an exemplary system according to one aspect of the invention. The system, generally identified with the numeral 10, commences with a sound source 12 that generates an input signal. The sound source 12 may be any apparatus for generating an audio signal for use in the invention, for example a microphone, optical disc player, magnetic tape player, RF receiver, computer system, etc. The sound source 12 may include internal processing of the signal it generates (e.g. amplification, normalization, bias adjustment, equalization, digital to analog conversion, noise reduction etc.) as is known in the art. Also, the sound source 12 may itself include a number of components that perform different functions, and a number of sound sources may together combine to provide the signal.

The sound source 12 is coupled to an equalizer 14. The equalizer 14 integrates the signal received from the sound source twice and then normalizes it so that it occupies an amplitude range of ± 1units. The double

integral is done to correct the second derivative effect of the demodulation process. The normalized signal from the equalizer 14 is then passed to a preprocessor 16, which is described in more detail below with reference to Fig. 2. The preprocessor 16 generates two signals, an "a" signal and a "b" signal, which are passed to a modulation block 18.

The modulation block 18, which will be described in more detail below with reference to Fig. 5, modulates the two signals onto two ultrasonic carrier waves that are 90° out of phase with each other. This is known as quadrature modulation. The ultrasonic carrier waves used to modulate the "a" and "b" signals have and identical frequency that is above the audible range of the human ear (e.g. above at least 15 kHz, normally above 20 kHz). The frequency of the carrier signals generated by the modulation block 14 may be any suitable frequency, and the frequency is typically selected so that all frequency components of the modulated signals are above 20 kHz. As an example only, a frequency of 40 kHz would be appropriate for use in the invention. After modulation, the two signals "a" and "b" are recombined in the modulation block 18, and they are passed on to a second equalizer 20. Equalizer 20 corrects for the distortion that occurs due to the bandwidth limitations of the ultrasonic emitters. The equalizer 20 has a transfer function that is an inverse transfer function of the transfer function of the ultrasonic emitters. This has the effect of equalizing the overall transfer function, thus increasing the overall bandwidth.

After leaving the equalizer 20, the signal is passed to a beamformer 22. The beamformer 22 is application-specific, and modifies the signal to generate the necessary signal(s) for beam formation and beam steering. The particular processing undertaken by the beamformer 22 is conventional in nature and will depend on the particular ultrasonic emitters used, and on the particular directional characteristics required. As such techniques are

known to those of ordinary skill in the art, they will not be discussed further here.

Also provided in the system 10 are an amplifier 24 and one or more ultrasonic emitters 26. The amplifier 24 provides amplification of the signal(s) received from the beamformer 22, and the ultrasonic emitters 26 transmit the amplified signal(s) into the air. The amplifier 24 and the ultrasonic transmitters 26 are conventional in nature, and their particular configurations (power levels, etc.) will depend on the particular application. An exemplary arrangement of ultrasonic emitters is shown in Fig. 4. As can be seen from the figure, a plurality of ultrasonic emitters 26 are mounted adjacent to one another on a shaped backing structure 27. The structure 27 is preferably paraboloid-shaped, which achieves better directivity than a traditional planar array. In addition, the directivity of the ultrasonic wave that is projected from the array can be controlled by adjusting the curvature of the array to achieve different focal points.

The preprocessor 16 is shown in more detail in Fig. 2. The signal received from the equalizer 14 is first received in a buffer 50. The preprocessor 16 processes the audio in a frame-by-frame manner, and the buffer 50 generates the frames to be processed by repeatedly allowing a selected period of audio to accumulate to form a complete frame, at which time the frame is passed on for further processing to an offset generator 52.

The offset generator 52 receives a frame from the buffer 50. As mentioned above, the signal has been normalized in the equalizer 14 to lie between a maximum value of +1 and a minimum value of –1 and to compensate for the second derivative demodulation effect in air. The offset generator 52 offsets the portion of the signal in the frame by an amount of +1. This ensures that no part of the signal in the frame is less than zero. This ensures that the subsequent square root operation can be performed on the entire frame and the results will all be real values. After performing

the offset, the offset generator 52 passes the offset signal to the square root module 54.

As the name implies, the square root module 54 takes the offset signal and performs a square root operation on all of the values of the offset signal. As mentioned above, the previous offset operation ensures that only real values result from the square root operation. After performing the square root operation, the square root module 54 passes the square root signal to a subtraction module 56. The subtraction module 56 also passes the signal on to the determining module 58.

One of the functions of the determining module 58 is to determine the smallest value of the square rooted signal in the frame. If the smallest value of the signal in the frame is within a tolerance (for example a value between 0 to 0.1), this value will be passed to the subtraction module 56, otherwise a value of 0 will be passed to the subtraction module 56. Passing a value of 0 to the subtraction module 56 will have the effect of not modifying the waveform. In the subtraction module 56, the received minimum value is subtracted from the entire portion of the signal in the frame, which has the effect of shifting the waveform down for non-zero received values. When the minimum value is subtracted, the lowest point(s) in the signal in the frame in question may now have a zero value. The shifted signal is then passed both to the "flipping point" determination module 58 and to the gain control module 60.

The determination module 58 determines where the turning points are in the shifted signal. This determination can be done by identifying where the slope (i.e. the first derivative) of the waveform goes from a negative value (a downslope from left to right) to a positive value (an upslope from left to right). In addition, these turning points have to be within a tolerance (for example a value of 0 to 0.1). This can be seen in Fig. 3, which shows an exemplary signal 100 received from the subtraction

module 56. In the figure, selected turning points can be seen at locations A and B, where the slope changes sign from negative to positive within the defined tolerance range. Also, as a result of the subtraction, it can be seen that in this example, the signal 100 has a zero value at one location, i.e. at A.

The determining module 58 then alternates the gain between +1 and −1 at each selected turning point. This has the effect of "flipping" the portion of the curve between every second set of selected turning points about the zero axis. The effect of this operation on the curve 100 in Fig 3 can been seen in the curve 110. Before the first selected turning point at A, the gain is set at +1 in this case. Note that the initial gain will always follow the gain of the last portion of the previous frame. The gain of +1 (or −1) is a multiplication factor, not an offset, so that the magnitude at any point on the curve remains unchanged. When the determining module 58 identifies a selected turning point at A, the determining module switches the gain to −1. The effect of this is to "flip" or "mirror" the portion of the curve 100 between points A and B about the zero axis. The result can be seen on curve 110 between points A and B. When the determining module 58 locates the next selected turning point on curve 100, at point B, the gain is again switched to +1. This results in the portion of curve 100 from point B onwards being identical to the corresponding portion of curve 110. In the illustrated example, there were only two selected turning points. If there had been additional turning points, the switching of the gain between +1 and −1 would have continued in the same manner at the additional selected turning points.

The determination of the turning points and the switching of the gain is done by determining module 58. The actual application of the gain is done by the gain control module 60. After leaving gain control modules 60, the signal in the processed frame now looks for example like curve 110 in

Fig. 3. This signal is passed to unbuffer 62, which is used to reassemble the frames before signal "a" leaves the preprocessor 16. Also, the gain control module also passes its output signal to the determining module 58 for use in generating signal "b."

In addition to determining the gain and the selected turning points, the determining module 58 also generates signal "b." Signal "b" is used to compensate for the difference in the resulting modulated signal between the ideal square root signal and signal "a." Also, since subsequent frames may have been subtracted by different values to generate signal "a", signal "b" is used to compensate for the discontinuities between frames in signal "a".

The determining module 58 takes the ideal square root signal (which was received from subtraction module 56 to enable the determination of the minimum value and the turning points) and subtracts the resulting envelope of signal "a" (for the frame in question) from that of the ideal square root signal. This compensates for the difference between the envelope of the ideal square rooted signal and envelope of signal "a" includes the subtraction of a different minimum value that may have been subtracted in each frame for the generation of signal "a." This compensation therefore takes into account the discontinuities between successive frames. The resulting frame-based signal "b" is passed to unbuffer 64, which functions in the same way as unbuffer 62, and from there to the modulation block 18. The resulting envelope of a signal $f(t)$, that is to be modulated by $\sin\omega_c t$ (i.e. the modulated signal is $f(t)*\sin\omega_c t$), can be easily found by taking the absolute value of $f(t)$ (i.e. $|f(t)|$).

Signals "a" and "b" are then passed to the modulation block 18. The modulation block 18 is shown in more detail in Fig. 5, and comprises two ultrasonic carrier wave generators 70, 72, two amplifiers 74, 76, and one adder 78. The ultrasonic carrier waves generate sinusoidal, ultrasonic waves onto which signals "a" and "b" are modulated. As can be seen from

the figure, generator 70 uses the cosine function of $\omega_c t$ to generate its carrier wave, while generator 72 uses the sine function of $\omega_c t$ to generate its carrier wave. That is, the carrier waves generated by the ultrasonic carriers wave generators 70, 72 are orthogonal to one another. It will of course be appreciated that other functions could be used to generate the ultrasonic carrier waves. The ultrasonic carrier waves thus generated are used to control the gain of amplifiers 74 and 76. The signals "a" and "b," which are respectively provided as inputs to the amplifiers 76, 74, are thus modulated onto the ultrasonic carrier waves. Modulated signals "a" and "b" are then provided to adder 78, where they are combined into signal "c." Signal "c" thus is defined by the equation $a.\sin(\omega_c t)+b.\cos(\omega_c t)$.

Signal "c" is then provided to the equalizer 20 as described above with reference to Fig. 1

Fig. 6 shows a comparison between the frequency spectrum of Blackstock's square rooted and modulated waveform and the modified square rooted and modulated waveform. In the figure, the audio signal is a combination of 1 kHz, 2 kHz and 4 kHz sinusoidal waves, while the ultrasonic carrier wave has a frequency of 40 kHz. As can be seen from the figure, the traditional square root method yields many harmonics, while the modified square root method of the invention does not. This has the advantage of reducing the bandwidth required of the ultrasonic emitters.

Fig. 7 shows the actual modulated signal (multi-tone signal of 1 kHz, 2 kHz and 4 kHz modulated onto 40 kHz carrier wave) for the traditional square root method and the modified square root method of the invention. As can be seen from the figures, there is little no difference between the envelope of the modulated signals provided to the ultrasonic emitters.

While this invention has been described in terms of several embodiments, it is contemplated that alterations, modifications and

permutations thereof will become apparent to those skilled in the art upon a reading of the specification and study of the drawings.

It is therefore intended that the following claims include all such alterations, modifications and permutations as fall within the spirit and scope of the present invention.

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CLAIMS

What is claimed is:

1. A method of processing an audio signal, comprising:

performing a square root operation on the audio signal to generate a square rooted signal;

alternating the gain of the square rooted signal between positive and negative gain values at selected locations to generate a flipped signal; and

modulating the flipped signal onto a first ultrasonic carrier wave.

2. The method of claim 1 further comprising the step of:

offsetting the audio signal by a predetermined amount prior to performing the square root operation to ensure that the square root operation only results in real values.

3. The method of claim 2 further comprising the step of:

dividing the audio signal into a plurality of frames;

determining, after the offsetting step; a minimum value of a portion of the audio signal in a particular frame; and

subtracting the minimum value from the portion of the audio signal in the particular frame.

4. The method of claim 3 further comprising the step of:

compensating the flipped signal in adjacent frames for discontinuities resulting from subtracting different minimum amounts in adjacent frames.

5. The method of claim 1 wherein the selected locations of the signal are minimum turning points of the signal.

6. The method of claim 1 further comprising the steps of:
determining a first modulation envelope for the flipped signal;
determining a second modulation envelope for the square rooted signal;
determining the difference between the first and second modulation envelopes;
modulating the difference between the first and second modulation envelopes onto a second ultrasonic carrier wave.

7. The method of claim 6 wherein the first and second ultrasonic carrier waves are orthogonal to one another.

8. The method of claim 4 further comprising the steps of:
determining a first modulation envelope for the flipped signal;
determining a second modulation envelope for the square rooted signal;
determining the difference between the first and second modulation envelopes;
modulating the difference between the first and second modulation envelopes onto a second ultrasonic carrier wave.

9. An apparatus for processing an audio signal received from an audio source, comprising:
a square root module to perform a square root operation on the audio signal to generate a
square rooted signal;
a determining module coupled to the square root module to alternate the gain of the square rooted signal between positive and negative gain values at selected locations, thereby to generate a flipped signal; and
a modulator to modulate the flipped signal onto a first ultrasonic carrier wave.

10. The apparatus of claim 9 further comprising:

an offset module to offset the audio signal by a predetermined amount prior to passing the signal to the square root module.

11. The system of claim 10 further comprising:

a buffer to divide the audio signal into a plurality of frames;

a subtracting module to subtract a minimum value from the portion of the audio signal in the particular frame.

12. The system of claim 9 wherein:

the determining module

determines a first modulation envelope for the flipped signal;

determines a second modulation envelope for the square rooted signal;

determines the difference between the first and second modulation envelopes;

the modulator;

modulates the difference between the first and second modulation envelopes onto a second ultrasonic carrier wave.

13. The apparatus of claim 9 wherein the selected locations of the signal are minimum turning points of the signal.

14. The apparatus of claim 12 wherein the first and second ultrasonic carrier waves are orthogonal to one another.

15. The apparatus of claim 13 wherein the turning points are below a selected minimum value.

16. A method for processing an audio signal received from an audio source, comprising:

processing the audio signal into a first processed audio signal;

processing the audio signal into a second processed audio signal;

modulating the first processed audio signal onto a first ultrasonic carrier wave; and

modulating the second processed audio signal onto a second ultrasonic carrier wave; wherein the first and second ultrasonic carrier waves have different phases.

17. The method of claim 16 wherein the first ultrasonic carrier wave is orthogonal to the second ultrasonic carrier wave.

18. The method of claim 17 wherein the step of processing the audio signal into the first processed audio signal comprises the step of:

applying a modified square root preprocessing method to the signal.

19. The method of claim 18 wherein the step of processing the audio signal into the second processed audio signal comprises the step of:

determining the difference between the envelope of an ideal square root signal and the envelope of the first processed audio signal.

20. The method of claim 19 wherein the modified square root preprocessing method comprises the steps of:

performing a square root operation on the audio signal to generate a square rooted signal; alternating the gain of the square rooted signal between positive and negative gain values

at selected locations to generate a flipped signal.

21. The method of claim 20 wherein the modified square root method comprises the further step of:

offsetting the audio signal by a predetermined amount prior to performing the

square root operation to ensure that the square root operation only results in real values.

22. The method of claim 21 wherein the selected locations of the signal are lower turning
points of the signal.

23. An apparatus for processing an audio signal received from an audio source, comprising:

a processor to process the audio signal into a first processed audio signal and a second
processed audio signal;

a modulator to modulate the first processed audio signal onto a first ultrasonic carrier wave and to modulate the second processed audio signal onto a second ultrasonic carrier wave; wherein the first and second ultrasonic carrier waves have different phases.

24. The apparatus of claim 23 wherein the first ultrasonic carrier wave is orthogonal to the second ultrasonic carrier wave.

25. The apparatus of claim 24 wherein the processor, when processing the audio signal into the first processed audio signal, applies a modified square root preprocessing method.

26. The apparatus of claim 25 wherein the processor, when processing the audio signal into the second processed audio signal, determines the difference between the envelope of an ideal square root signal and the envelope of the first processed audio signal.

27. The apparatus of claim 26 wherein the modified square root preprocessing method applied by the processor comprises the steps of:

performing a square root operation on the audio signal to generate a square rooted signal; alternating the gain of the square rooted signal between positive and negative gain values

at selected locations to generate a flipped signal.

28. The apparatus of claim 26 wherein the modified square root method preprocessing method applied by the processor comprises the further step of:

offsetting the audio signal by a predetermined amount prior to performing the square root operation to ensure that the square root operation only results in real values.

29. The apparatus of claim 27 wherein the selected locations of the signal are lower turning points of the signal.

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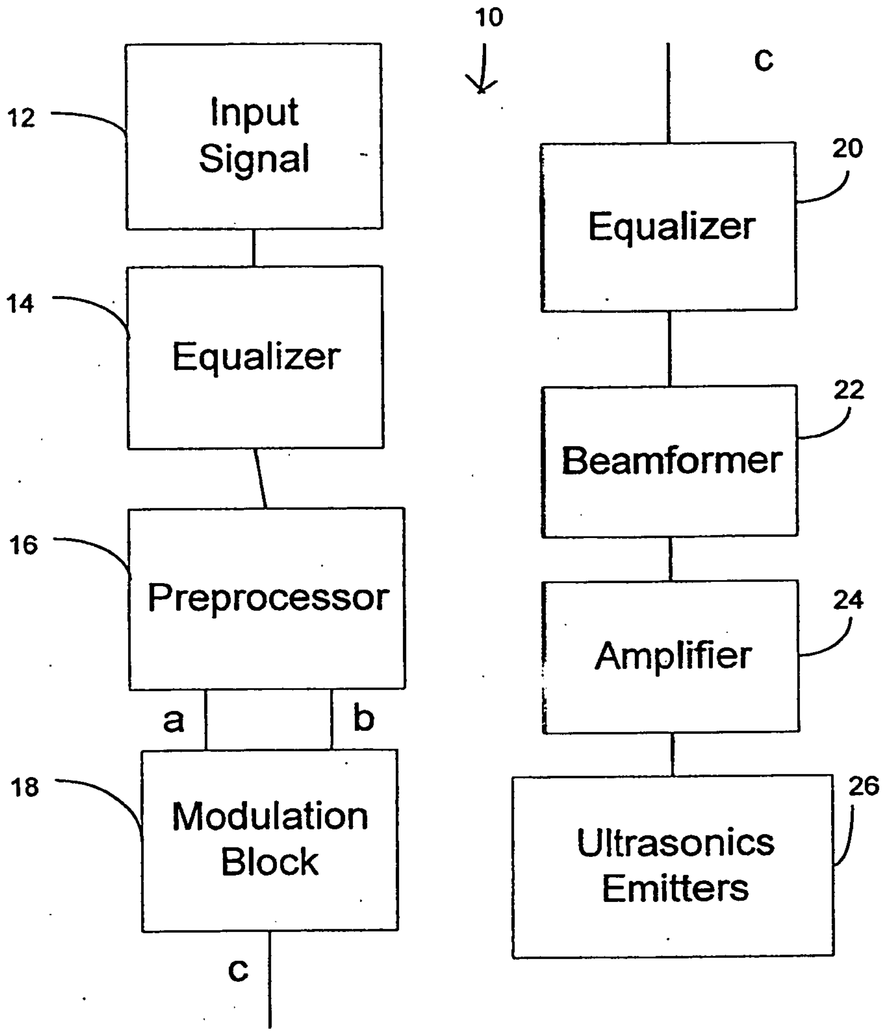

FIG. 1

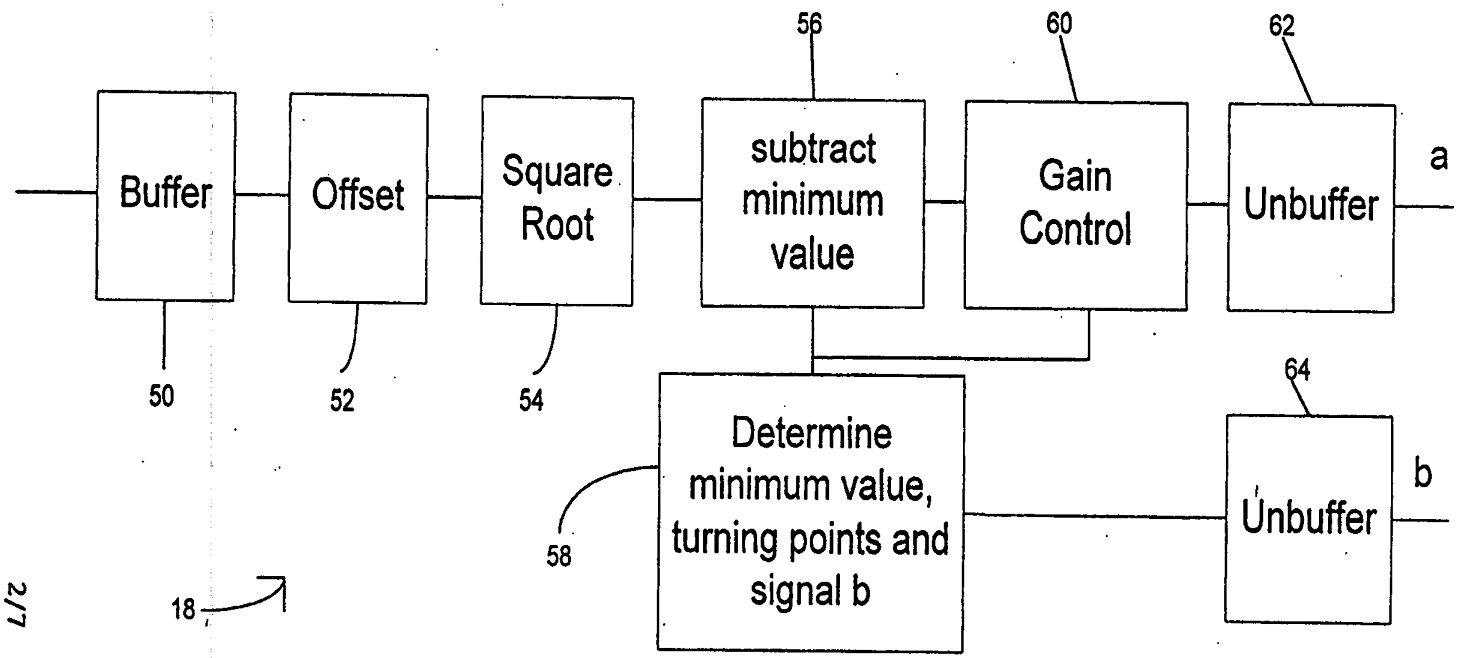

FIG. 2

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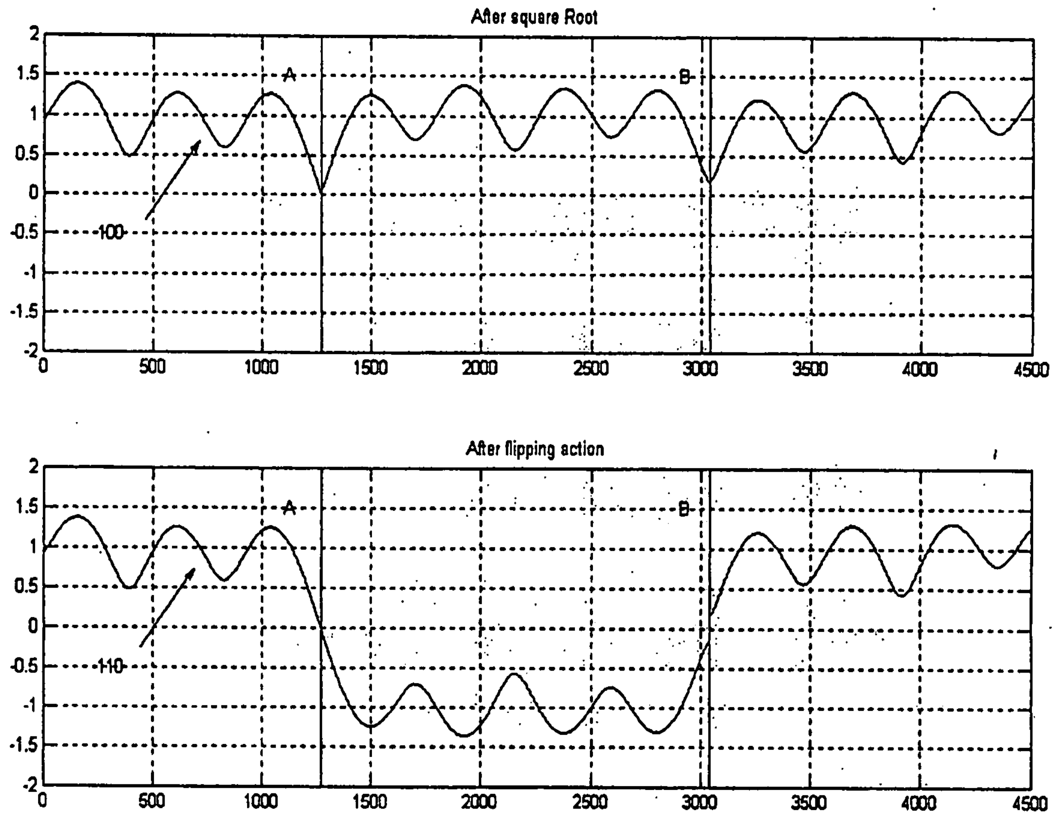

FIG. 3

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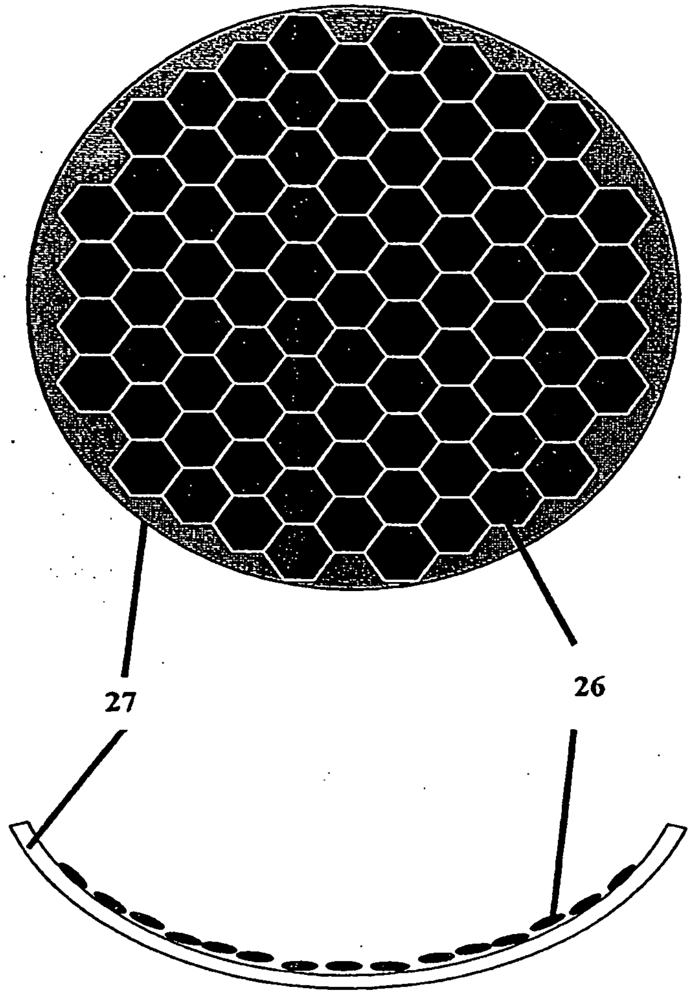

FIG. 4

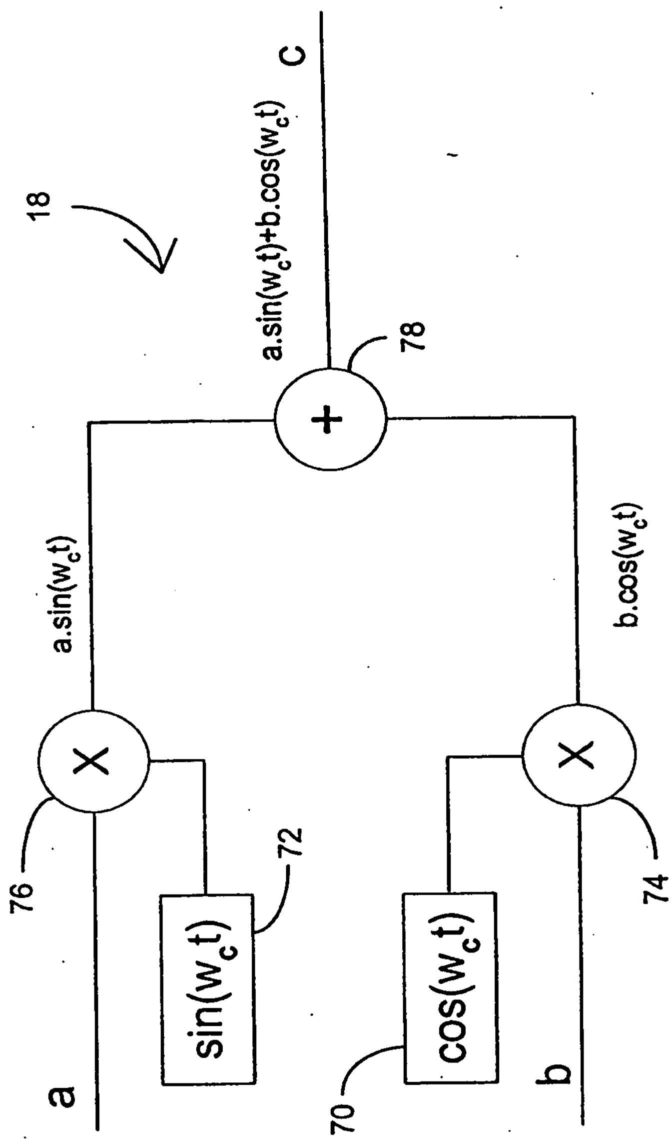
18
a
b
c
sin(w_c t)
cos(w_c t)
a.sin(w_c t)
b.cos(w_c t)
a.sin(w_c t)+b.cos(w_c t)
70
72
74
76
78

FIG. 5

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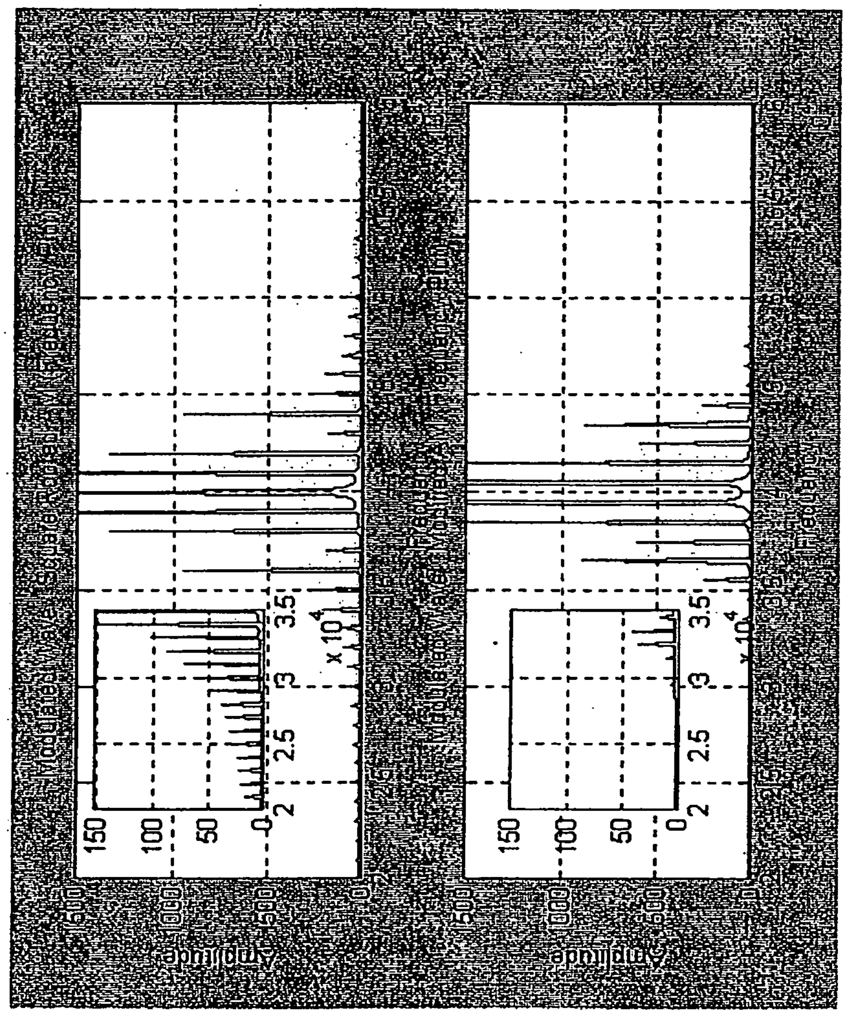

FIG. 6

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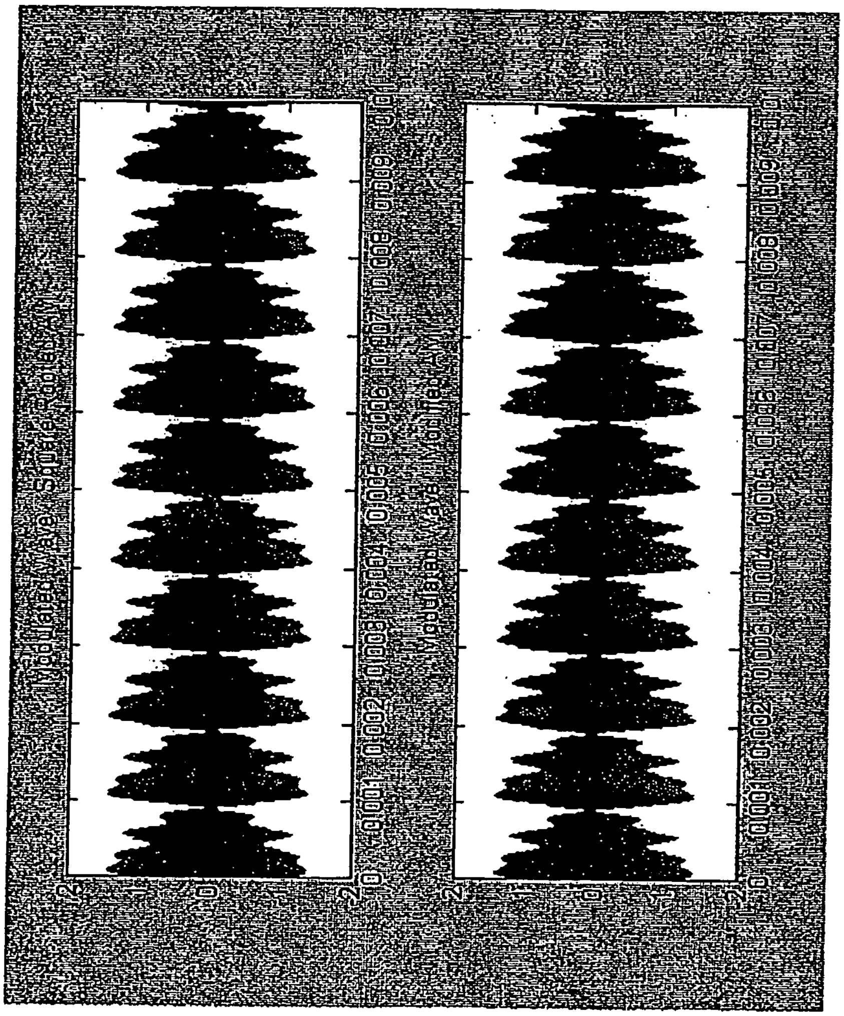

FIG. 7